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EXAMINER	
OPSASNICK, MICHAEL N	
ART UNIT	**PAPER NUMBER**
2655	

DATE MAILED: 07/13/2005

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)	
	09/782,791	GAO, YANG	
	Examiner	Art Unit	
	Michael N. Opsasnick	2655	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *05 May 2005*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-13,15-47 and 49-70* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-13,15-47,49-70* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. Claims 1-5, 12,13,15-20, 27-34, 42-47, 49, 54-58, 63-66 are rejected under 35 U.S.C. 103(a) as being unpatentable over Ertem et al. (6,453,289) in view of Borth et al (4630304).

As per claims 1 and 45, Ertem et al. (6,453,289) disclose:

A full-rate encoder (36, FIG. 3 and Col. 3, line 57). Inherently, the encoder is capable of providing a bit stream based on the type of speech coding. Ertem et al. disclose a pre-compression mode of noise-reduction (FIG. 1 and Col. 3, lines 43-49);

ACELP coder (Col. 4, lines 5-7) which inherently determines at least one gain based on the encoding (from gain codebook). In addition, Ertem et al. teach that their noise-reduction algorithm will operate on essentially all coders (Col. 3, lines 62-66);

Encoder adjusting gain based as a function of noise characteristic (gain functions are computed using smoothed noise spectral estimate - Col. 11, lines 59-61).

As per claims 1,45, Ertem et al. (6,453,289) does not explicitly teach adjusting the gain factor using background noise attenuation wherein the signal is a/d converted, then the signal is time domain to frequency domain converted, apply the background noise attenuation, and then reconvert back to the time domain, however, Borth et al (4630304) teaches a noise reduction system focusing on adjusting the gain according to the noise floor and noise estimates by performing a/d, time to frequency conversion, background noise estimation and removal, and conversion back to the time domain (Borth et al (4630304), col. 3 line 35 – col. 4 line 2). Therefore, it would have been obvious to one of ordinary skill in the art of noise reduction systems to modify the teachings of Ertem et al. (6,453,289) with gain adapting noise modification in the time to frequency back to time domain because it would advantageously remove high levels of background ambient noise (Borth et al (4630304), col. 1 lines 14-24).

As per claims 2 and 46, Ertem et al. (6,453,289) disclose the use of CELP coder (Col. 4, lines 5-7).

As per claim 4, Ertem et al. (6,453,289) disclose adjusting gain prior to quantization (precompression mode, Col 4, lines 15-25).

As per claims 5, 20, 49, 58, Ertem et al. (6,453,289) discloses adjusting gain (g_agc) by a constant Beta (gain factor) (Equation 3, Col. 6, lines 50-55).

As per claims 12 and 54, Ertem et al. (6,453,289) disclose a full-frame coder (Col. 3, line 5657), although the system can operate with other frame rates. (Col. 3, lines 62-66).

As per claim 13, Ertem et al's. (6,453,289) system uses a DSP chip for noise suppression and encoding. (Col. 3, lines 60-61) The noise-reducing portion of the nGER31/PC board containing a DSP chip inherently receives digital signal from the AID converter (which receives and converts analog signal from the microphone) and modifies spectral magnitudes of the digitized signal (elems. 94, 97, FIG. 7).

As per claim 15, Ertem et al. (6,453,289) disclose a decoder (elem. 24, FIG. 1).

As per claims 16 and 55, Ertem et al. (6,453,289) disclose:

A decoder (28, FIG. 2) performing noise-reduction in post-compression mode (FIG. 2 and Col. 3, lines 43-49).

ACELP decoder (Col. 4, lines 5-7) which inherently determines at least one gain based on the decoding (from gain codebook).

Decoder adjusting gain based as a function of noise characteristic (gain functions are computed using smoothed noise spectral estimate - Col. 11, lines 59-61).

As per claims 16,55, Ertem et al. (6,453,289) does not explicitly teach adjusting the gain factor using background noise attenuation wherein the signal is a/d converted, then the signal is time domain to frequency domain converted, apply the background noise attenuation, and then reconvert back to the time domain, however, Borth et al (4630304) teaches a noise reduction system focusing on adjusting the gain according to the noise floor and noise estimates by performing a/d, time to frequency conversion, background noise estimation and removal, and conversion back to the time domain (Borth et al (4630304), col. 3 line 35 – col. 4 line 2). Therefore, it would have been obvious to one of ordinary skill in the art of noise reduction systems to modify the teachings of Ertem et al. (6,453,289) with gain adapting noise modification in the time to frequency back to time domain because it would advantageously remove high levels of background ambient noise (Borth et al (4630304), col. 1 lines 14-24)..

As per claims 17 and 56, Ertem et al. (6,453,289) disclose the use of CELP coder (Col. 4, lines 5-7). Inherently, CELP coder requires CELP decoder for decompression.

As per claim 19, Ertem et al. (6,453,289) disclose adjusting gain after decoding (post-compression mode, Col 4, lines 25-35).

As per claim 27, Ertem et al. (6,453,289) disclose a full-frame coder/decoder (Col. 3, line 5657), although the system can operate with other frame rates. (Col. 3, lines 62-66).

As per claim 28, Ertem et al's. (6,453,289) system uses a DSP chip for noise-suppression and encoding (Col. 3, lines 60-61). Inherently, a similar DSP chip will be required for decoding.

As per claim 29, Ertem et al. (6,453,289) disclose a decoder (elem. 26, FIG. 2).

As per claims 3, 18, 47, 57, Ertem et al. (6,453,289) disclose using ACELP, which is a variation of CELP (Col. 4, lines 5-7) in the compression coding art. Ertem et al. do not disclose using eX-CELP, which is another variation of CELP coder in the compression coding art.

Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Ertem et al. (6,453,289) to use eX-CELP instead of ACELP. Applicant has not disclosed that ex-CELP provides an advantage, is used for a particular purpose or solves a stated problem. One of ordinary skill in the art, furthermore, would have expected Ertem et al.'s apparatus to perform equally well with either ACELP or eXCELP because both of these coders are art recognized equivalent variations of standard CELP coding, for the purpose of providing compression coding.

As per claims 30 and 63, Ertem et al. (6,453,289) disclose:

A full-rate encoder (36, FIG. 3 and Col. 3, line 57). Inherently, the encoder is capable of providing a bit stream based on the type of speech coding. Ertem et al. disclose a pre-compression mode of noise-reduction (FIG. 1 and Col. 3, lines 43-49)

ACELP coder (Col. 4, lines 5-7) which inherently determines at least one gain based on the encoding (from gain codebook). In addition, Ertem et al. teach that their noise-reduction algorithm will operate on essentially all coders (Col. 3, lines 62-66)

Encoder adjusting gain based as a function of noise characteristic (gain functions are computed using smoothed noise spectral estimate - Col. 11,lines 59-61)

A decoder (28, FIG. 2) performing noise-reduction in post-compression mode (FIG. 2 and Col. 3, lines 43-49)

ACELP decoder (Col. 4, lines 5-7) which inherently determines at least one gain based on the decoding (from gain codebook).

Decoder adjusting gain based as a function of noise characteristic (gain functions are computed using smoothed noise spectral estimate - Col. 11, lines 59-61).

As per claims 30,63, Ertem et al. (6,453,289) does not explicitly teach adjusting the gain factor using background noise attenuation wherein the signal is a/d converted, then the signal is time domain to frequency domain converted, apply the background noise attenuation, and then reconvert back to the time domain, however, Borth et al (4630304) teaches a noise reduction system focusing on adjusting the gain according to the noise floor and noise estimates by performing a/d, time to frequency conversion, background noise estimation and removal, and conversion back to the time domain (Borth et al (4630304), col. 3 line 35 – col. 4 line 2). Therefore, it would have been obvious to one of ordinary skill in the art of noise reduction systems to modify the teachings of Ertem et al. (6,453,289) with gain adapting noise modification in the time to frequency back to time domain because it would advantageously

remove high levels of background ambient noise (Borth et al (4630304), col. 1 lines 14-24).. Furthermore, as per claims 30 and 63, the examiner takes official notice that it is well-known in the art of speech processing to reduce noise both before encoding and after decoding, in order to minimize the total amount of noise in the signal. By reducing the noise before the signal is encoded, the system ensures that it transmits cleaner signal and by post-processing the signal after decoding, the system removes the noise added during transmission. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Ertem et al. (6,453,289) to use noise attenuation before the signal has been encoded (encoder with noise filter) and after the signal has been decoded (decoder with noise filter) in order to reduce the overall level of noise in the signal. This would allow the system to encode cleaner signal and also remove the noise added during transmission at the time of decoding (Col. 4, lines 15-26). In other words, this would improve system performance as the system would attenuate noise on both transmission and reception sides, thus reducing the amount of noise more effectively than a system with a single noise-attenuation component.

As per claims 32, 65, the combination of Ertem et al (6,453,289) in view of Borth et al (4630304) does not disclose using eX-CELP, which is another variation of CELP coder in the compression coding art. Ertem et al. (6,453,289) disclose using ACELP, which is a variation of CELP (Col. 4, lines 57) in the compression coding art. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify Ertem et al. to use eX-CELP instead of ACELP. Applicant has not disclosed that ex-CELP provides an advantage, is used for a particular purpose or solves a stated problem. One of ordinary skill in the art,

furthermore, would have expected Ertem et al.'s (6,453,289) apparatus to perform equally well with either ACELP or eXCELP because both of these coders are art recognized equivalent variations of standard CELP coding, for the purpose of providing compression coding.

As per claims 31 and 64, Ertem et al. disclose the use of CELP coder/decoder (Col. 4, lines 5-7).

As per claim 33, Ertem et al. disclose adjusting gain prior to quantization (precompression mode, Col 4, lines 15-25)

As per claims 34 and 66, Ertem et al. discloses adjusting gain (g agc) by a constant Beta (gain factor) (Equation 3, Col. 6, lines 50-55).

As per claim 42, Ertem et al. disclose a full-frame coder (Col. 3, line 56-57), although the system can operate with other frame rates. (Col. 3, lines 62-66)

As per claims 43-44, Ertem et al.'s system uses a DSP chip for noise suppression and encoding. (Col. 3, lines 60-61) The noise-reducing portion of the nGER31/PC board containing a DSP chip inherently receives digital signal from the A/D converter (which receives and converts analog signal from the microphone) and modifies spectral magnitudes of the digitized signal (elems. 94, 97, FIG. 7).

3. Claims 6-11, 21-26, 35-41, 50-53, 59-62, 67-70, are rejected under 35 U.S.C.

103(a) as being unpatentable over the combination of Ertem et al. (6,453,289) in view of Borth et al (4630304) in further view of Chandran et al. (6,671,667).

The respective base claims 5,20,34,49,58,66 are unpatentable over the combination of Ertem et al. (6,453,289) in view of Borth et al (4630304) for the reasons listed above in the 103 rejection.

As per claims 6-7, 21-22, 35-37, 50, 59, 67, the combination of Ertem et al. (6,453,289) in view of Borth et al (4630304) does not disclose a variable gain factor G=1-C*NSR, where C is in the range of 0 to 1 or 0.4 to 0.6. However, Chandran et al. (6,671,667) teaches computing gain factor using formula: G(n) = 1-Wn*NSR, where W is a weighting factor. (Eq. 1, Col. 6, lines 30-40). In addition, the gain factor must lie within [0,1] range (Col. 6, lines 36-40), which is only possible if Wn*NSR is always less or equal to 1. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the combination of Ertem et al. (6,453,289) in view of Borth et al (4630304) as taught by Chandran et al. (6,671,667) in order to improve the noise attenuating performance of the system, since Chandran et al.'s gain function calculation allows to limit the gain when system has noisy input (high NSR). Neither the combination of Ertem et al. (6,453,289) in view of Borth et al (4630304) nor Chandran et al. (6,671,667). teach keeping W within [0,1] range or even further, keeping it in [0.4,0.6] range. However, G should increase (amplification) when NSR is low (speech signal) and decrease (attenuation) when NSR is high (noise signal). Because NSR itself is limited to [0,1] range, W must not exceed 1 when NSR approaches 1 for G to lie within [0, 1] range. The value of W is less relevant when signal is mostly speech (NSR is low). Therefore, it would have

been obvious to one of ordinary skill in the art at the time the invention was made to the combination of Ertem et al. (6,453,289) in view of Borth et al (4630304) in further view of Chandran et al. (6,671,667) to keep W within [0,1] range in order to keep overall gain factor less responsive to the changes, when NSR is high, since in these situations, making W greater than 0 would cause rapid fluctuations in G(n) (it will become negative and possible very large), thus causing undesirable attenuation of the signal. Furthermore, it would have been further obvious to one of ordinary skill in the art at the time the invention was made to modify the combination of Ertem et al. (6,453,289) in view of Borth et al (4630304) in further view of Chandran et al. (6,671,667) to keep W within [0.4, 0.6] range for performance reasons, as this would provide even further stability of Gain factor's values, since at this range of W, Gain function would not be adversely affected by sudden increases of NSR values and also would keep some level of noise in the signal, making the speech sound more realistic.

As per claims 8, 23 and 38, Ertem et al. (6,453,289) discloses VAD (elem. 32, FIG. 3).

As per claims 9-11, 24-26, 39-41, 51-53, 60-62, 68-70, the combination of Ertem et al. (6,453,289) in view of Borth et al (4630304) teaches computing a variable gain faction (abstract). Chandran et al. teach computing a variable gain function and smoothing it using weights (Eq. 1). Neither Ertem et al. nor Chandran et al. teach computing gain faction based on the running mean. The examiner takes the official notice that it is extremely well-known in the art to compute running means of the variable, when the smoothed, time-averaged value of the

variable is required in order to avoid sharp fluctuations in variable values. The running mean formula is well-known is usually of the form: $X(i+1) = bX(i-1) + (1-b)X(i)$, where $0<=b<1$. Therefore, it would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the combination of Ertem et al. (6,453,289) in view of Borth et al (4630304) in further view of Chandran et al. (6,671,667) to use variable gain function based on the running mean, in order to improve the performance of the system by smoothing the value of the gain function and thus reducing the possibility of sudden fluctuation in gain function values.

Response to Arguments

4. Applicant's arguments with respect to the newly amended claim language have been considered but are moot in view of the new ground(s) of rejection.

Conclusion

5. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Please see related art listed on the PTO-892 form.

6. **Any response to this action should be mailed to:**

Commissioner of Patents and Trademarks
Washington, D.C. 20231
or faxed to:
(703) 872 9314,

(for informal or draft communications, please label "PROPOSED" or "DRAFT")

Hand-delivered responses should be brought to Crystal Park II, 2121 Crystal Drive, Arlington. VA., Sixth Floor (Receptionist).

7. Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael Opsasnick, telephone number (571)272-7623, who is available Tuesday-Thursday, 9am-4pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Mr. Wayne Young, can be reached at (571)272-7582. The facsimile phone number for this group is (571)272-7629.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the Group 2600 receptionist whose telephone number is (571) 272-2600, the 2600 Customer Service telephone number is (571)272-2600.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

mno
7/2/05

Michael N. Opsasnick
Examiner
Art Unit 2655